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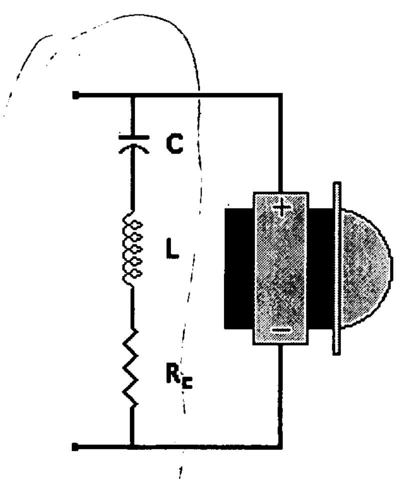
Series Notch Filter Designer Help

Series Notch Filters are designed to dampen driver resonance at its resonance frequency (fs). Drivers can produce 10db (or more) peaks at their resonance frequency, and this filter will remove them. This is used mostly in mids and tweeters. Usually with a woofer, large inductors are required which are costly and and have a large resistance which causes its own problems. If using a ferrofluid (or some other magnetic oil) cooled tweeter, then the resonance should already be damped and the filter is probably not necesssary. Also, if the crossover point is 2 octaves or more above the resonance frequency, and a 2nd order or greater crossover is used, then the filter is probably not necessary.

A Series Notch Filter is simply a capacitor (C), inductor (L) and resistor (Rc) all in series, in parallel with the driver. Sometimes, a series notch filter is called a LCR filter, because of the L, C, and R components.



Re = Driver DC Resistance in Ohms
fs = Driver Resonance Frequency in Hz

Qes = Driver Electrical Q
Qms = Driver Mechanical Q

$$C = \frac{.1592}{Re * Qes * fs}$$

$$L = \frac{.1592 * Qes * Re}{fs}$$

$$Rc = Re + \frac{Qes * Re}{Qms}$$

Use these formulas if you dont know Qes & Qms

$$C = \frac{.03003}{fs}$$

$$L = \frac{.02252}{fs^2 * C}$$

$$Rc = Re$$

When using the second set of formulas, experiment with different values of Re in .5 Ohm increments. This assumes you can measure frequency response. Otherwise, try to use a driver that you know the Q values for. If designed properly, a series notch filter usually does not require testing or experimenting. It should work properly on the first try.

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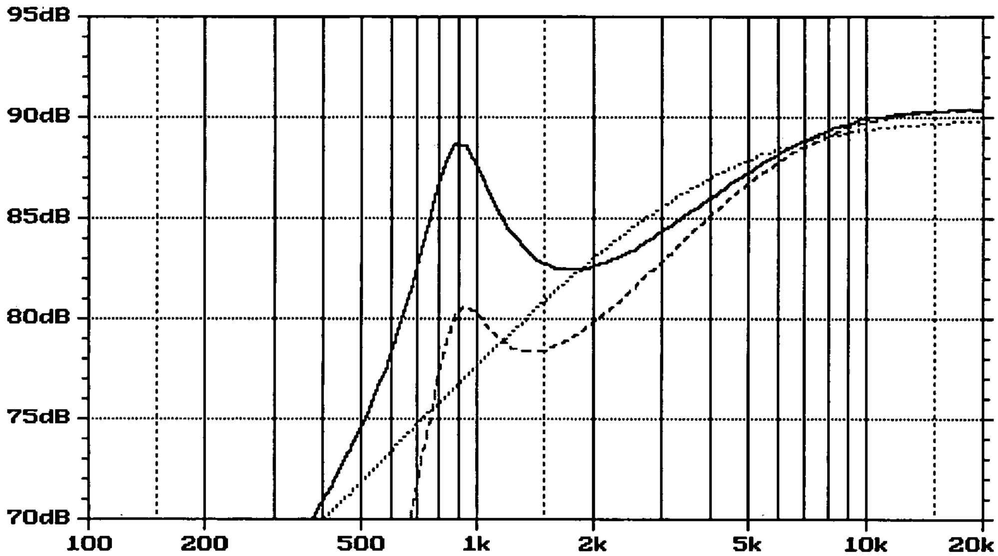
Example: Say you have a tweeter with a Re of 8 Ohms a fs of 1000 Hz, and want to use a first order (6db/octave) Butterworth high pass crossover at 4000 Hz. The tweeter does not use magnetic oil cooling, and the crossover point is two octaves from the resonance frequency. The driver resonance will affect the frequency response as seen below. The first order crossover is just a 5 uF capacitor in series with the driver.

Actual driver response with no crossover

Desired driver response

Actual Driver response with crossover



With and without the crossover, there is a large spike in frequency response a the crossover point. Using a fs of 1000 Hz and a Re of 8 Ohms, with the second set of formulas you get C = 30.03 uF, L = .75 mH and Rc = 8 Ohms.

Think of the capacitor as a HPF and the inductor as a LPF. This creates a small bandpass circuit where the current will flow through the resistor. Outside of these bandpass frequencies, the resistance of the series notch filter and the driver is simply the resistance of the driver. In the notch, the resistance is much higher, depending on how much you want to dampen the resonance.

Impedance curve of the series notch filter only

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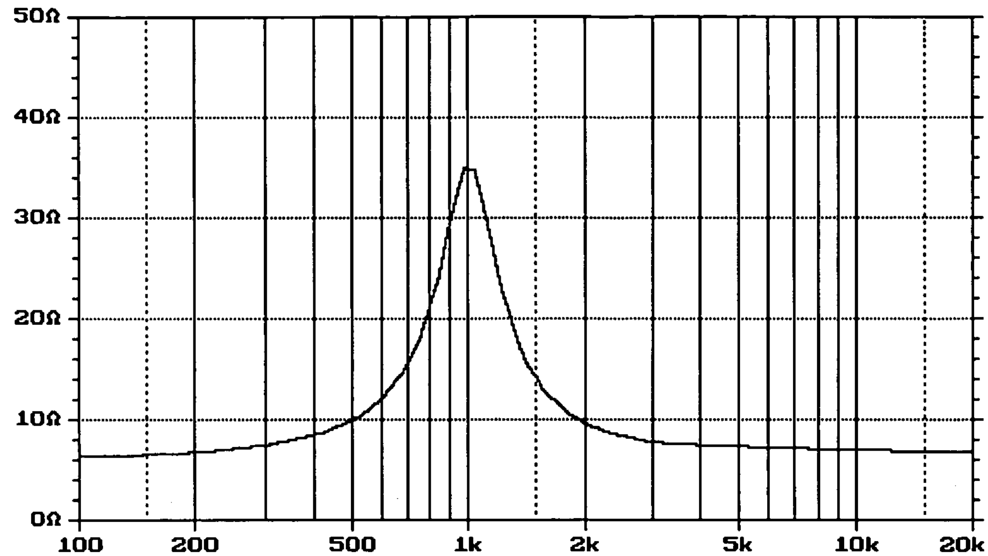
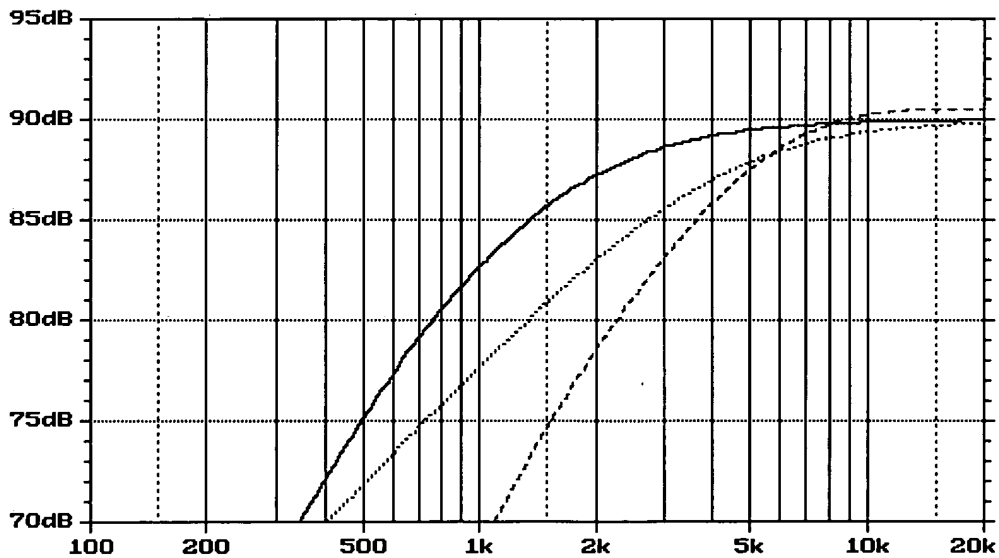


When the series notch filter is used with the driver, you get the frequency responce shown below. The reason the combined crossover & series notch filter response curve is below the desired response is because the rolloff of the driver and crossover are summed.

Actual driver response with the series notch filter

Desired driver response

Actual Driver response with the crossover and the series notch filter



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h e c c c e e ch he h

Carnegie Mellon

Control Tutorials for Matlab

Notch Filter

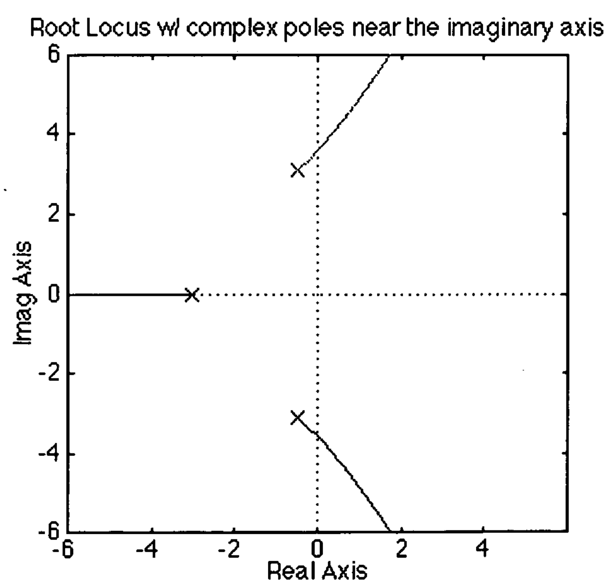
There are many times when the transfer function of a controlled process contains one or more pairs of complex-conjugate poles that lie close to the imaginary axis in the s-plane. This will result in an undesirable closed-loop system that is unstable or only lightly damped. For example, consider the following transfer function,

$$G_p(s) = \frac{K}{(s+3)(s^2+s+10)}$$

If you were to take the root locus of this system,

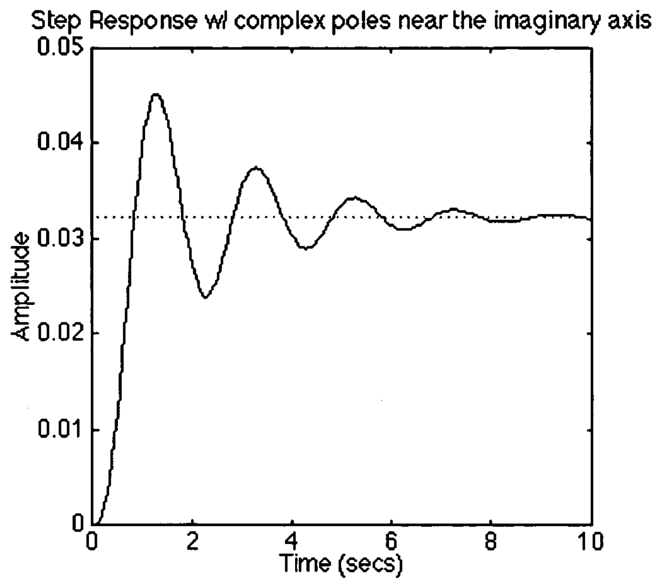
```
num = 1;
den = conv([1 3],[1 1 10]);
rlocus(num,den)
```

you would end up with the following plot



As you can see, the plot shows that this system is only stable for a small region of the root locus. The portion that is stable will only be lightly damped (small zeta). By closing the loop and plotting the step response to this system in the portion of the root locus that is in the right half plane (assuming a gain of one), you can see that the response is poor.

```
[numc,denc]=cloop(num,den,-1);
step(numc,denc)
```



There is a large overshoot, long settling time, and a large steady-state error. If you try to increase the gain, you will see that the response improves slightly, but becomes unstable before a desirable response can achieved. Pure proportional control is obviously not a good way to control this system.

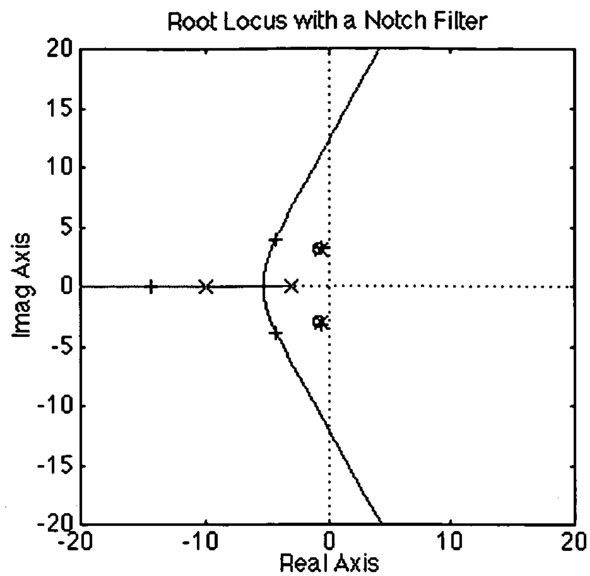
One way to control this system is to design a controller with zeros near the undesirable, lightly-damped poles of the plant. These zeros can attenuate the effect of these poles. The poles of the controller can then be placed in a more desirable position. Such a controller is called a **notch filter**. Before getting into the specifics of a notch filter, it should be noted that due to the nature of most systems, exact pole/zero cancellation cannot be obtained; nor should it be attempted. Approximate cancellation will give us many of the desirable characteristics without the pitfalls.

For the example above, let's try placing the zeros slightly to the left of the lightly-damped poles (it is a good idea to pull the poles to the left instead of the right). Try the following compensator

$$G_c(s)=\frac{s^2+1.5s+10}{s^2+20s+100}$$

As you can see, the roots of the numerator of the controller are almost the same as the complex poles of the denominator of the plant; the denominator of the controller introduces two poles at -10. If we implement this controller into our m-file and plot the root locus

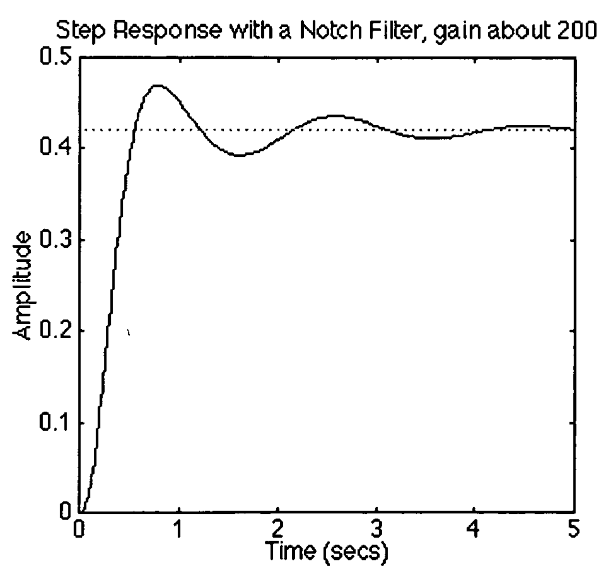
```
numnotch = [1 1.5 10];
dennotch = [1 20 100];
newnum = conv(num,numnotch);
newden = conv(den,dennotch);
rlocus(newnum, newden);
```



The complex poles near the imaginary axis have been nearly canceled and more of the root locus is now in the left half plane. This means that higher gains, k, can be used, while maintaining stability. Furthermore, higher damping can be achieved using lower gains (zeta term will be larger). If we close this loop (using the `rlocfind` command to find the gain), and look at the step response, the system response should look much better.

```
[k,poles]=rlocfind(newnum,newden)
[numc,denc] = cloop(k*newnum,newden,-1);
step(numc,denc)
```

When prompted to select a point, pick a point near one of the white crosses in the root locus above. The gain (k) should be about 200.



This is a much better step response. The overshoot and settling time are both smaller. There is still a steady-state error, but that can be reduced with a lag controller later. The placement of the poles in the controller is a matter of trial and error. They should be sufficiently to the left to pull the root locus into the left half plane, but their exact location can be determined based on the desired root locus plot and by

trial and error.

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8/29/96 JDP

The Subwoofer DIY Page v1.1
Bandpass Systems: Notch Filter

last updated: 6th May1998

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Notch Filter

Your bandpass design may suffer from out of band noise, unless you take some steps to reduce it. One method of doing this is to use a passive Parallel Notch Circuit in series with the driver to get rid of the noise. This filter circuit looks like the following:

```
                  C
          +------||-------+
          |       L       |
o---------+---~~~~~~~~----+---------o
          |       R       |
          +---/\/\/\/\----+
```

To calculate the values of C, R, and L, you'll first need to find F, the midpoint of the noiseband. Also find F1 and F2, the frequencies at which the response drops by 3dB compared to the response at F. Alternatively, assume that the majority of the noise problem occurs at the pipe resonance frequency of the port (344/(2*L)), where L is the length of the port in metres, and also assume 0.95*F and 1.05*F as the -3dB frequencies F1 and F2.

Then:

```
C = 0.03003/F (Farads)
L = 0.02252/(F^2*C) (Henries)
R = 1/(6.2832*C*(F1-F2)) (Ohms)
```

Note that capacitors are usually rated in microfarads (uF) and coils in milliHenries (mH). To convert:

```
1 F = 1,000,000 uF
1 H = 1000 mH
```

Example:

Center of noise band, F, is at 300 Hz -3dB points occur at 250 Hz and 360 Hz

```
C = 0.03003/F
  = 0.03003/300
  = 0.0001001 Farads

L = 0.02252/(F^2*C)
  = 0.02252/(90000*0.0001001)
  = 0.00248 Henries

R = 1/(6.2832*C*(F1-F2)
  = 1/(6.2832*.0001001*(360-250))
  = 1/(6.2832*0.0001001*110)
  = 14.5 Ohms
```

Values required (closest match)

C = 100 uF (bipolar)

H = 2.50 mH (preferably air-core)
R = 14.5 ohms

```
power rating for R is
   approx = Pe*R/(R+Re)

where,
Pe = speaker power rating
Re = speaker's min impedance
```

A Notch Filter

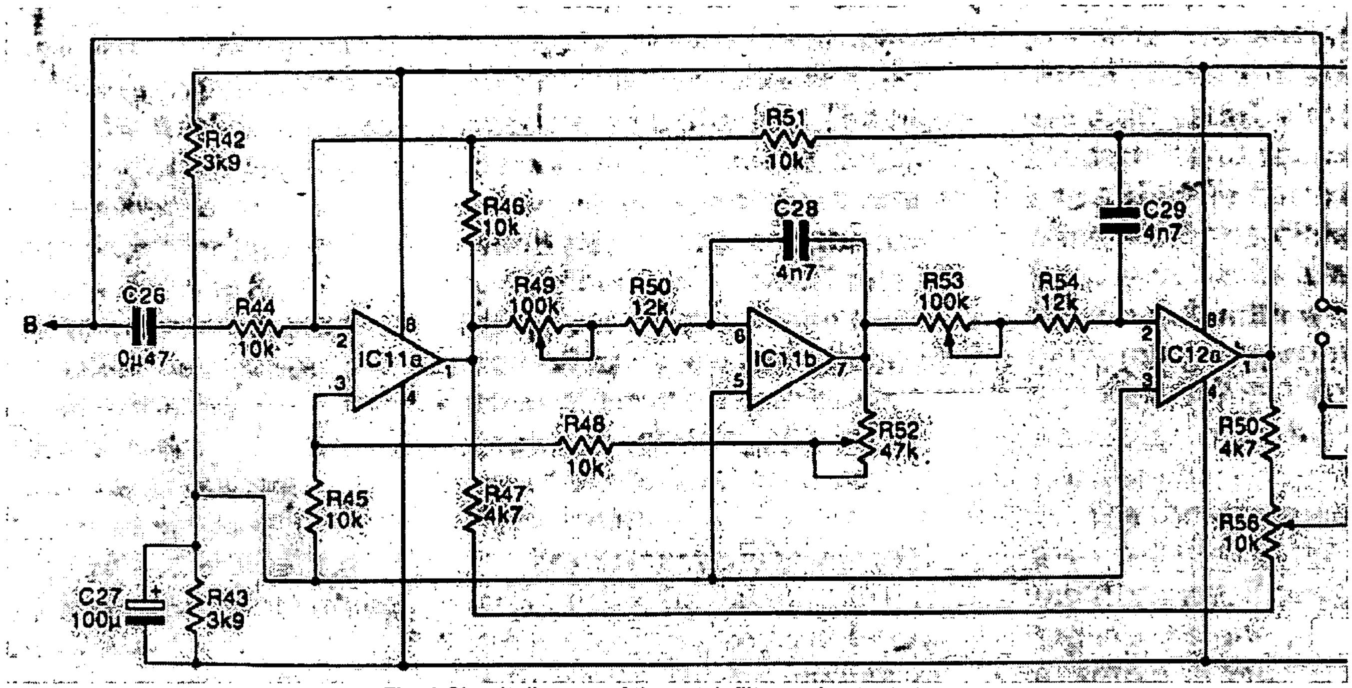

Fig. 1 Circuit diagram of the notch filter and output stage.

The circuit diagram of a notch filter and output stage appears in Fig. 1. The notch filter is based on IC11 and IC12, which are used in a conventional state variable filter. In this case it is only the notch output at pin 7 of IC12 that .is utilized. Ganged potentiometers R49 and R53 form the tuning control, and R56 is the balance control. The latter is adjusted to optimize the attenuation at the centre of the notch, and around 40dB to 60dB of attenuation can be achieved.

The resistor R52 enables the Q of the filter to be varied. A high Q value gives a relatively narrow notch, with little attenuation well away from the notch frequency. This type of response is best for dealing with heterodyne tones, as it enables the tone to be dealt with effectively while leaving the wanted signal largely intact. A Low Q gives a somewhat wider notch, and also tends to give significant losses well away from the notch frequency. This obviously has a more detrimental affect on the main signal, but it is more effective at combating interference that covers a small range of frequencies (such as an RTTY signal with a shift of 170Hz).

Switch S1 enables the notch filter to be bypassed when it is not required. From S1 the signal is coupled to an attenuator and then to the output amplifier. This amplifier is based on an LM386N-1, which requires few discrete components. In this case it is operated at minimum gain, and the only discrete components required are a DC blocking capacitor at the output (C32) and a supply decoupling capacitor (C33). Resistor R59 is used to reduce the maximum drive current to a level that suits most headphones. For sensitive headphones a higher value of about 390 will give better results. If the unit is used with insensitive headphones, or to drive a loudspeaker having an impedance of 8D or more, R59 should be replaced with a shorting link.

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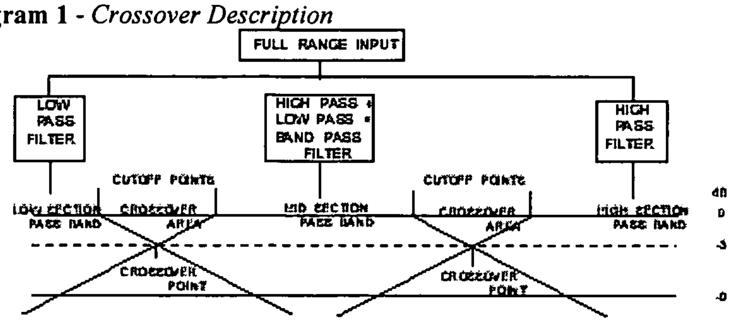
Tech Support
Crossovers & Filters

A crossover is a component or a group of components which limit the frequencies that reach a speaker. If . separates. (subwoofers, woofers, midranges and/or tweeters) are used in a system, they should not receive a frequency range greater than their design limits. Crossovers would be utilized to limit the frequency range to each separate.

Crossover Points

The frequency range which a filter allows full power to pass through it is called the passband. The frequency at which a filter begins reducing power is referred to as the cutoff point. The range between two adjoining cutoff points is the crossover area.

Diagram 1 - *Crossover Description*



As the frequencies move away from the passband area (starting at the cutoff point) power reduction increases. When it has reached 3dB in reduction, the crossover point has been reached.

All crossovers and filters are rated at their crossover point (or crossover frequency). This minus 3dB point is also known as the half power point.

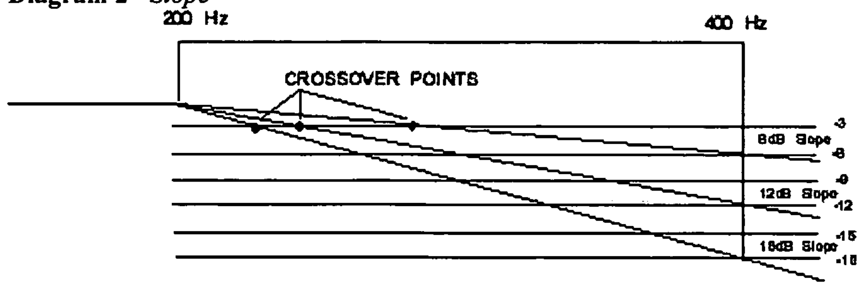
There are two main types of crossover filters, the high pass, and the low pass. The high pass will pass high frequencies while restricting low frequencies. The low pass does just the opposite, in that it passes low frequencies while restricting high frequencies.

By combining a high pass and a low pass circuit in series, we create a band pass circuit, which allows a certain band of frequencies to pass, while restricting both high and low frequencies.

Slopes

Crossover filters may be constructed with different rates of reduction of power. Slope is the term used to describe this rate.

Diagram 2 - *Slope*



A 6dB per octave filter will reduce power by 6 dB in every octave starting at its cutoff point. For instance, a 6dB per octave low pass filter with a cutoff of 200 Hz would reduce power 9dB (6+3) at 400 Hz (200 times 2). At the end of the next octave, 800 Hz, an additional 6dB of power reduction would have been reached.

A 12dB per octave filter would have twice as much reduction per octave. An 18dB per octave filter would have three times as much reduction as a 6dB per octave filter.

If all frequency bands (low, mid, high) are at the same acoustical level, and the speakers for each section are receiving a frequency range well within their response capabilities, the slope of the crossover filter is not of great importance. On the other hand, if one section is at a much higher level than its adjoining section, or the range directed to a speaker is very close to its maximum range capabilities, or a great deal of power is received by a speaker, a steeper slope may be useful.

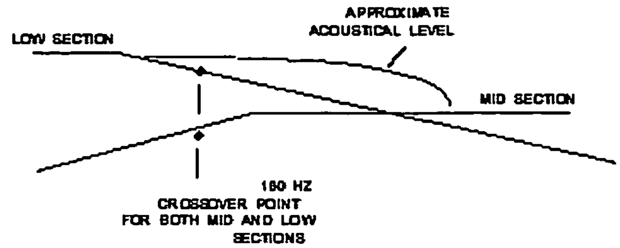
Diagram 3 shows a low frequency section which is much louder than the adjoining. This will occur when a much higher level is used for the low section than the mid section speaker.

The result is a bulge or broad peak in the combined acoustical output of the two sections at and above the crossover area. The crossover point, the slope, or both may be changed to correct the bulge.

A 6dB per octave filter will reduce power by 6 dB in every octave starting at its cutoff point. For instance, a 6dB per octave low pass filter with a cutoff of 200 Hz would reduce power 9dB (6+3) at 400 Hz (200 times 2). At the end of the next octave, 800 Hz, an additional 6dB of power reduction would have been reached.

A 12dB per octave filter would have twice as much reduction per octave. An 18dB per octave filter would have three times as much reduction as a 6dB per octave filter.

Diagram 3 - *Acoustical imbalance*



If a crossover point is close to the end of a speaker. s response range, the speaker may distort or not perform well in or near the crossover area. This problem can become critical as more power is applied or when more power is initially used in a system. A steeper slope may solve this problem by reducing the power of lower frequencies more quickly. A faster rate of reduction (12dB per octave or higher) is used when a crossover frequency change is not desired.

As an example, a small midrange speaker may not perform well below 200 Hz, but for staging reasons, you do not wish to raise the woofers above 200 Hz. A faster rate of reduction with the high pass filter for the midrange may enable it to operate with a crossover point of 200 Hz.

The amount of power applied to a midrange (or tweeter) may shorten its useable range, especially low frequency response. As power is increased, the practical range of the speaker will compress. Consequently, where the same crossover points are used with 50 watts of power, they may not be acceptable at 100 watts of power. Changing crossover points and/or increasing the filter slope will enable the speaker to perform well without distortion.

6dB per octave passive filters are commonly used in mobile audio systems and operate quite satisfactorily. Crossover points should be well within the range of the speakers used. In addition, they cost half as much as 12dB per octave passives and one-third as much as 18dB per octave passive filters.

With tweeters, an 18dB per octave passive filter is advantageous as its greater slope adds some protection to this small and delicate speaker.

Electronic Crossovers

An electronic crossover operates at preamp level to limit the frequencies to the amplifier or amplifiers connected to it. The speakers which are connected to the amplifier(s) would therefore receive a limited frequency range.

It is a powered electronic circuit which limits or divides frequencies. The circuit operates independently and does not care about speaker impedance nor does it create any appreciable signal loss.

With active crossovers, the filters are built onto its circuit board. Changing the filters (or crossover points) is accomplished through external dial turning, by changing frequency modules with a switch or by changing crossovers if fixed types are used.

Active crossovers are usually 12dB per octave. There are some available which are 18dB per octave and higher. Generally, 12dB per octave is fine.

A crossover which has no crossover frequency adjustment is usually termed "Fixed Frequency". Where crossover frequency changes are possible through a switch, knob or module change, they are called "adjustable or variable". The advantage of fixed is cost; of variables, the ability to change crossover points

with only one crossover model.

An adjustable crossover allows the user to make crossover changes easily and to immediately hear the effect of the changes.

Most electronic crossovers have output controls for each individual channel. This allows you to set the gains for all amplifiers at one convenient location, as well as the ability to level match a system.

Some crossovers will allow you to set the low and high pass filters separately, which allows you to tune out. acoustic peaks or valleys at or near the crossover frequencies.

One of the advantages of electronic crossovers is that there is little or no insertion loss. Passives reduce the amplifier power slightly, due to their resistance.

Another advantage of electronic crossovers is the ability to separate low frequencies into their own exclusive amplifier, which reduces distortion heard at high volumes in the high frequency speakers. Amplification of low frequencies requires greater power than higher frequencies. When an amplifier is at or near peak output, clipping may occur, which is able to destroy tweeters and other speakers with small voice coils. A separate low frequency amplifier allows the total system to play louder and with lower distortion.

Passive Crossovers
A passive crossover filter is made exclusively with coils and/or capacitors. Passive filters are changed by using different coils and capacitors to obtain the crossover point desired.

A 6dB per octave low pass is simply a coil in series on the speaker lead. A 12dB per octave is a coil in series and a capacitor which is in parallel with the speaker. An 18dB per octave low pass filter is a coil in series, followed by a capacitor in parallel and another coil (of different value) in series.

A high pass circuit can be created in the same manner, except that wherever there is a coil, there is a capacitor and wherever there is a capacitor, there is a coil.

Diagrams 4,5 and 6 show the various low pass circuits.
Diagrams 7,8 and 9 show the various high pass circuits.

Diagram 4 - *6 dB Low Pass*

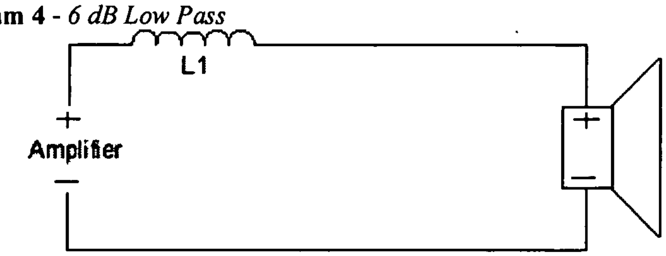


Diagram 5 - *12 dB Low Pass*

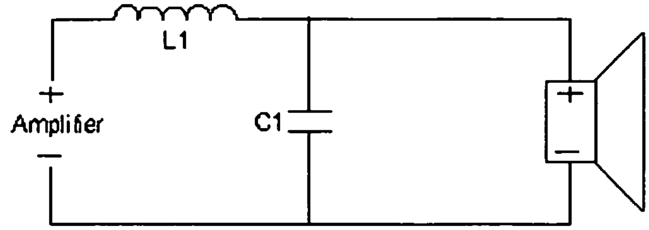


Diagram 6 - *18 dB Low Pass*

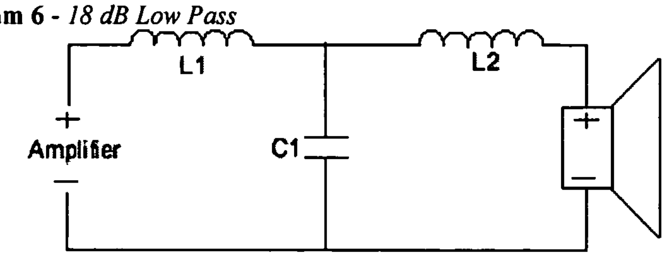


Diagram 7 - *6 dB High Pass*

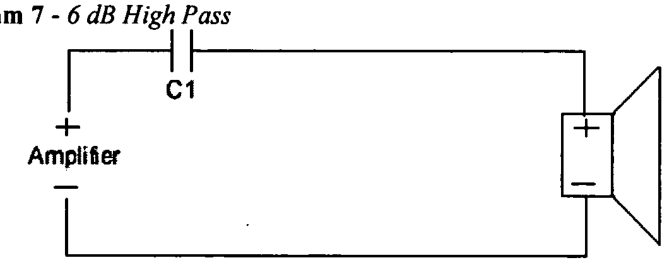


Diagram 8 - *12 dB High Pass*

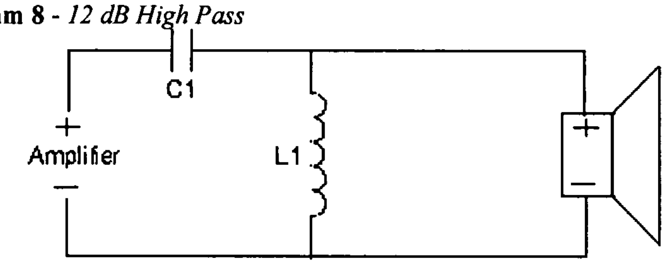
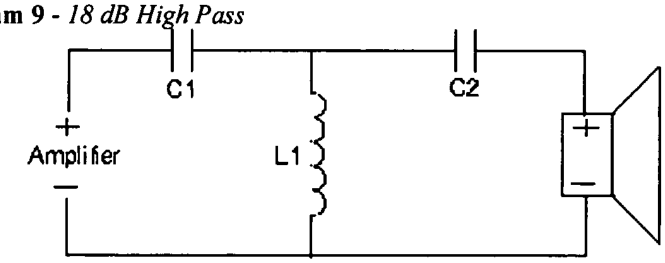


Diagram 9 - *18 dB High Pass*



Band pass filters are a high pass filter and a low pass filter in series. Diagrams 10, 11, and 12 show 6,12 and 18dB per octave band pass filters.

Diagram 10 - *6 dB Band Pass*

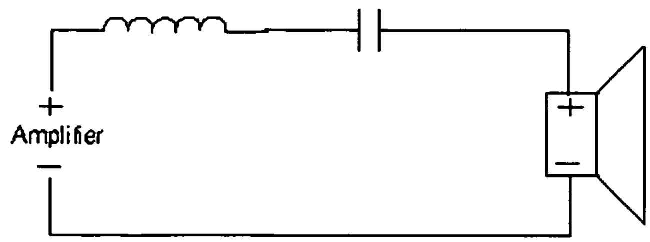


Diagram 11 - *12 dB Band Pass*

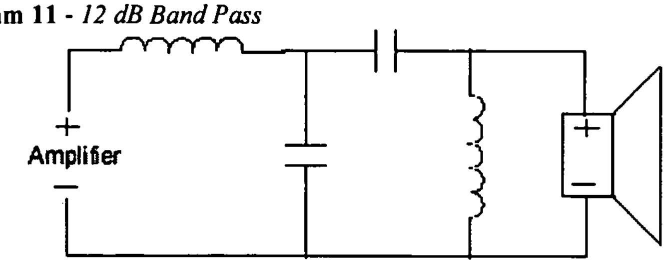
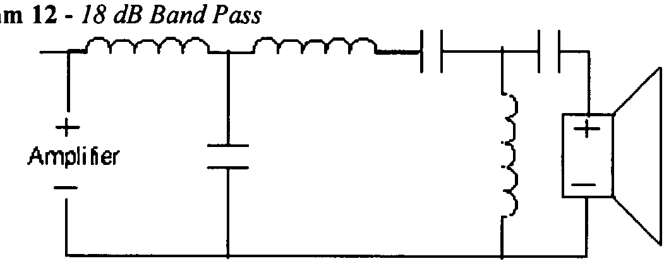


Diagram 12 - *18 dB Band Pass*



One concept that is commonly misunderstood about crossovers, is that along with changing the acoustical response, the impedance is affected as well. What this means is that you can have a 4 ohm tweeter and a 4 ohm woofer, use a proper crossover, and create a 4 ohm load. It is very easy to make the mistake by thinking that it would create a 2 ohm load because of the drivers being in parallel.

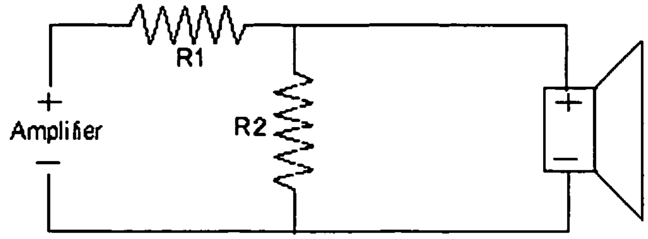
Attenuation

High frequency drivers are commonly more efficient than low frequency drivers. This creates a need to adjust the driver levels to create a uniform overall frequency response.

L-pads, which are the type described here, are the most suitable for adding to crossover circuits, because they do not change the resistive load shown to the crossover.

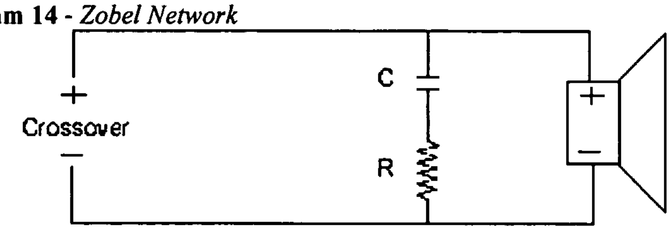
Diagram 13 depicts the wiring of an attenuation network.

Diagram 13 - *Attenuation Circuit*



Zobel All voice coil type drivers exhibit a rising impedance caused by the voice coil inductive reactance. In order for crossover circuits to operate properly, you can equalize this rising impedance by using the CR circuit shown in Diagram 14. You can also use this circuit for tweeter domes, not to facilitate network operation, but to help eliminate harshness, and to assure the accurate application of L-type shelving networks.

Diagram 14 - *Zobel Network*



Notch Filter The primary function of the circuit illustrated in Diagram 15 is to dampen and eliminate the effects of driver resonance on crossover networks. Assuming the driver has an undamped resonance peak, and the peak is located less than two octaves from a high pass crosspoint, this circuit will greatly improve driver performance. It is particularly useful on tweeter domes, midrange domes, and cone type midrange drivers whose enclosure resonance is above 200 Hz.

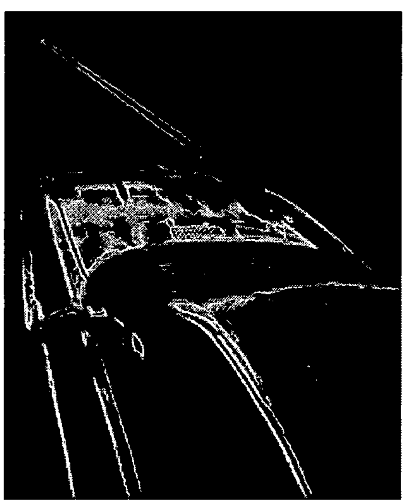
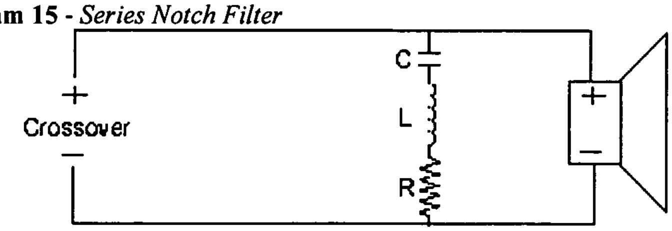


Diagram 15 - *Series Notch Filter*



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